Amendments to the Specification

This listing of following amended paragraphs will replace all prior versions, and listings, of the specification:

Please amend the title as follows:

~~METHOD AND PROCESSOR SYSTEM FOR PROCESSING OF AN AUDIO SIGNAL~~

MINIMIZING RESOURCE CONSUMPTION FOR SPEECH RECOGNITION PROCESSING WITH DUAL ACCESS BUFFERING

Please amend the paragraph at page 1, line 23 as follows:

The ~~letter~~ latter aspect is of particular importance for controlling portable devices, such as mobile phones, personal digital assistants or palm top computers.

Please amend the paragraph at page 4, line 2 as follows:

On the other hand there is a variety of applications where the usage of such high performance standard microprocessors is not desirable for a number of reasons. Firstly, the ~~prize~~ price of adding an additional processor for the speech recognition can be unacceptably high. Secondly, in the case of portable electronic devices the power consumption of an additional high performance processor can drastically reduce battery lifetime.

Please amend the paragraph at page 6, line 16, as follows:

Further the core processor 16 is coupled to dual access stack 8. The dual access stack 8 is coupled to the back-end processor 21. The back-end processor 21 is connected to a non-volatile memory 22 which stores the program to be executed by the back-end processor 21. Further the back-end processor 21 is connected to a digital to analog converter 23 to supply an analog signal to the amplifier 24. The output of the amplifier [[14]] 24 provides an analog audio signal which is supplied to a speaker.

Please amend the paragraph at page 6, line 25 as follows:

Further the back-end processor 21 is coupled to gain control [[24]] 25 for controlling the gain of the amplifier 24.

Please amend the paragraph at page 8, line 3 as follows:

In response to this the program stored in the memory 17 of the core processor 16 is started and the data which are buffered in the dual access stack 7 are read for further processing by the core processor 16. In parallel the streaming of data from the front-end processor 12 to the dual access stack 7 continues. The point of time when the trigger signal is sent out by the front-end processor can be a predetermined time interval after the first clock ~~poles~~ pulse of the clock C1 supplied by line 30. Alternatively this point of time can be determined by the data volume which has been output from the front-end processor 12 to the dual access stack 7.

Please amend the paragraph at page 10, line 9 as follows:

The acoustic modeling together with the Viterbi search are representing the most performance demanding section of the entire voice recognition algorithm requiring processing performance of above 100 MIPS. This indicated performance demand is valid for today's voice recognition algorithms for example defined for [[a]] an active vocabulary of 500 words. An increase of the vocabulary will extend the demand on performance even more.